MediaPack™ Series Analog Gateways
Release Notes
Version 4.2
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Abbreviations

Each abbreviation, unless widely used, is spelled out in full when first used. Only industry-standard terms are used throughout this manual. Hexadecimal notation is indicated by 0x preceding the number.

Applicable Products

This manual provides additional information for the following AudioCodes products:

MediaPack Series: MP-124/FXS, 24 analog FXS interfaces.

MP-108/FXS, 8 analog FXS interfaces.

MP-108/FXO, 8 analog FXO interfaces.

MP-104/FXS, 4 analog FXS interfaces.

MP-104/FXO, 4 analog FXO interfaces.

MP-102/FXS, 2 analog FXS interfaces.

Note: With this manual, as CD, Web site or other electronic copies, all cross-references are hyperlinked, so just click on the page or section numbers to reach the individual cross-referenced item directly.

Notice

These Release Notes describe the functionality of the AudioCodes' MediaPack Series Boards supported by Software Release 4.2. Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, AudioCodes cannot guarantee the accuracy of printed material after the Date Published nor can it accept responsibility for errors or omissions.

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This document is subject to change without notice.

Please refer to the current release notes that may be included with your documentation or hardware delivery.

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Reader's Notes

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Reader's Notes

1 What's New in Release 4.2

1.1 Hardware Supported in Release 4.2

1. This version supports the following MediaPack hardware platforms (no change from previous release):

   - MediaPack MP-124/FXS, 24 analog FXS interfaces
   - MediaPack MP-108/FXS, 8 analog FXS interfaces
   - MediaPack MP-108/FXO, 8 analog FXO interfaces.
   - MediaPack MP-104/FXS, 4 analog FXS interfaces
   - MediaPack MP-104/FXO. 4 analog FXO interfaces.
   - MediaPack MP-102/FXS, 2 analog FXS interfaces

1.2 Analog Interface New Features

2. MINIMUMFLASHHOOKTIME *ini* file parameter. Relevant for FXS only. Sets the minimal time (in msec) for detection of a flash hook event. Flash hooks that last a shorter period are ignored.

3. Polarity reversal detection. Relevant only for FXO. Polarity reversal is a reverse in the Tip-Ring voltage. Two consecutive polarity reversals within a time interval above 350 msec are detected each as a single reversal polarity. An event acEV_ANALOG_IF_REVERSAL_POLARITY_DETECTED is issued for each detection.

4. Wink detection. Relevant for FXO only. A wink is generated by two line polarity reversals within a time interval of 100 msec up to 350 msec. An event acEV_ANALOG_IF_WINK_DETECTION is issued upon detection of a wink. This event is available also in MGCP.

5. Metering tone detection. Relevant for FXO only. Ability to detect metering tone signals of 12 kHz and 16 kHz (each frequency requires a separate coefficients file). On detection of a metering tone event acEV_ANALOG_IF_DETECT_METERING_TONE is issued. This event is also supported in MGCP.

6. Current disconnect detection. Relevant for FXO only. This feature enables the FXO to detect a current disconnection (also known as loop disconnect) of 800-1000 msec. A new event is issued - acEV_ANALOG_IF_CURRENT_DISCONNECT_DETECTED.

7. Line disconnect detection. Relevant for FXO only. This feature enables the FXO to detect a physical analog line disconnection when the current disconnect is longer then 1000 msec. A new event is being issued - acEV_ANALOG_IF_LINE_DISCONNECTED.

8. Line connect detection. This feature enables the FXO to detect a physical analog line connection when current is restored. a new event is being issued - acEV_ANALOG_IF_LINE_CONNECTED.

9. FarEndDisconnect detection. Relevant for FXO only. This new mechanism allows the MP to detect a far end disconnection based on several criteria: detection of certain call progress tones and/or polarity reversal and/or current disconnect. A new event - acEV_FAR_END_DISCONNECTED is introduced (also supported in H.248 and MGCP).

10. FarEndConnect detection. This detection is being made by the FXO upon a polarity reversal

event. A new event is issued: acEV_FAR_END_CONNECTED.

11. NTT CallerID type one generation. Relevant for the FXS only. This feature enables the FXS to transmit the call originator's telephone number to the end point connected to the FXS.

12. NTT CallerID type one detection. Relevant for the FXO only. This feature enables the FXO to display the call originator's telephone number while the FXO is the terminator.

1.3 Host API New Features

13. The following new events were added:

    acEV_FAR_END_CONNECTED

    acEV_FAR_END_DISCONNECTED

    acEV_ANALOG_IF_WINK_DETECTION

    acEV_ANALOG_IF_DETECT_METERING_TONE

    acEV_ANALOG_NTT_CALLER_ID_FAILED

    acEV_ANALOG_IF_CURRENT_DISCONNECT_DETECTED

    acEV_ANALOG_IF_LINE_DISCONNECTED

    acEV_ANALOG_IF_LINE_CONNECTED

14. The following new parameters / definitions were added:

    - acTDuslicDistinctiveRingParams was added to BoardParam->AnalogIFFXSSettings.

15. The version includes the following bug fix:

    - A problem with acAnalogIFGenerateDistinctiveRing() was fixed.

16. For additional changes which are not MP-specific refer to AudioCodes.

1.4 IP Networking Related New Features

17. Improved NAT. Two new NAT board parameters were introduced into the NAT feature: EnableIpAddrTranslation and EnableUdpPortTranslation. These parameters allow the User to specify the type of compare operation that takes place on the first incoming packet. The User can specify, for example, that when receiving a first incoming packet with different source UDP port, NAT does take place.

18. Robust reception of RTP streams using a new filtering mechanism. This new mechanism filters out unwanted RTP streams that are sent to the same port on the board. These multiple RTP streams can result from traces of previous calls, call control errors and deliberate attacks.

19. New API for online network configuration - acChangeNetworkParameters(). This new API allows the User to change the IP address, subnet mask and default Gateway address without resetting the board. This API doesn't affect currently opened channels - only channels that are opened after this API is invoked behave according to the new network parameters.

20. ICMP unreachable packet reporting - the board can now report on reception of ICMP unreachable packets via acEV_REPORT_ICMP_UNREACHABLE (andacEV_REPORT_ICMP_REACHABLE). The User can configure the mechanism using the parameters EnableICMPUnReachableReport and ICMPUnReachableReportInterval.

21. LAN watchdog - This new mechanism detects LAN failures on the board and restarts the board if required. A LAN failure can result from a software or hardware malfunction. A new board parameter EnableLanWatchDog enables this feature (default is disabled).

22. Updated HeartBeat configuration. The 4-byte Heart Beat payload carried over the Heart Beat header was replaced with the sender's MAC address. The Payload Type was updated to 0x1. The sending interval range was extended to allow long intervals (in terms of days).

1.5 VoIP Engine New Features

23. Jitter Buffer parameters DJBufMinDelay & DJBufOptFactor can now be changed on-the-fly using the acChangeChannelParameters function. Upon such a change, the Jitter Buffer will reset before applying the new settings.

24. Support for reception of RTP packets with the header Padding bit set to 1.

25. Support for different RFC 2833 payload type values for sending and receiving using 2 new parameters: RFC2833RTPTxPayloadType and RFC2833RTPRxPayloadType.

26. Support for dialing of DTMF and MF digits to the IP network over RFC 2833 (previously only in-band dialing was supported).

27. Improved Broken Connection event mechanism – User can now set the event timeout using the BrokenConnectionEventTimeout parameter (in 100 msec units). In addition, BrokenConnectionEventActivationMode controls if the detector is activated when the channel is opened, or from first RTP packet received from the network.

28. Adding Packets and Octets received fields to the acEV_RTCP and acEV_RTCP_CLOSE events.

29. Supporting Cisco™ NSE mode for Modem automatic pass-through.

30. Fax/Modem bypass Basic payload size control. With the new FaxModemBypassBasicRTPPacketInterval channel parameter, the User can send multiples of 5 msec and 10 msec RTP frame size during Fax/Modem bypass session.

31. Addition of a new field called CallerIDGenerationStatus at the Caller ID ended event. This field reports if the Caller ID generation finished successfully or not.

32. Activate/Deactivate T.38 API functions [acActivateT38Channel(), acDeActivateT38Channel()] were added to enable better control over T.38 sessions

33. Support for reordered RTP packets – a new algorithm has been implemented to handle RTP packets which are received reordered. This feature improves the voice quality on a network which suffers from packet reordering problems.

34. Configurable Answer-Detector Sensitivity support. This can reduce the false alarm rate.

35. CNG fax tone detection and T.38 Relay support. Enables fast fax session detection for CNG-generating fax machines.

36. Support for a special Jitter Buffer optimization mode for Modem or Fax. User should set the jitter buffer Optimization Factor to 13 for Modem or Fax calls.

Note: Installation and use of vocoders is subject to obtaining the appropriate license in advance of use and to royalty payments

37. The DSPVersionTemplateNumber parameter selects the DSP firmware and hence the supported coder group. The following list specifies the different values for DSPVersionTemplateNumber and their respective supported coder list:

    - 0 = DSP firmware supports PCM/ADPCM, G.723, G.729A.
    - 1 = DSP firmware supports PCM/ADPCM, (this Template provides backward-compatibility).
    - 2* = DSP firmware supports PCM/ADPCM, G.723, G.729A.
    - 3* = DSP firmware supports PCM/ADPCM, (this Template provides backward-compatibility).

(* These templates include additional media processing features: Answer Detector and longer Echo Canceler: 40 msec.)

1.6 MGCP New Features

38. New package:

    - FMTP package - dynamic payload type

39. New signals:

    - Call waiting2 (wt2)
    - Call waiting3 (wt3)
    - Call waiting4 (wt4)
    - Reversal polarity generation
    - Wink signal (prp)
    - Metering tone (mtr)

40. New event:

    - RTP/RTCP Timeout (rto)
    - Reversal polarity detection
    - Network disconnect (osi)

41. Support for second endpoint id according to RFC 2705

42. Improved MGCP Quarantine mode

43. Increase max number of Digit Maps to 50.

44. Gateway now sends new RSIP if call agent does not accept the endpoint with “200 OK” response

45. Support O (origin), T (time) and S (session) fields in the SDP section of the MGCP response.

46. Keep-alive heartbeat implemented using a proprietary RSIP message which is sent every

configurable interval.

1.7 MEGACO New Features

47. The following MEGACO (H.248) package was added in this version:

    - FXO Package (fxo)

48. The retransmission mechanism timers can be configured by the following *ini* file parameters:

    - MGCPRetransmissionTimeout (or MGCPRetransmitionTimeout) - Defines the initial value (in milliseconds) for re-sending a transaction if answer did not arrive.

    - MGCPCommunicationLayerTimeout – Defines the expiration time (in seconds) for the whole retransmission session. If this time expires, the MEGACO enters a disconnection mode.

49. New *ini* file parameter - KeepAliveInterval for setting the interval between two KeepAlive transactions.

50. The outgoing SDP can contain now the 't' 's' 'o' lines, which are mandatory in some none-MEGACO applications. This option is controlled by the existing *ini* file parameter 'MGCPCompatibilityProfile', by adding to the current value (2) the number 8. So if you need the SDP to have these lines, you should set the *ini* file parameter to 10.

51. Support of T.38 was added through the SDP. Refer to the User's manual for exact details.

52. The use of RFC 2833 can be negotiated via SDP. Refer to the User's manual for more details.

53. The use of silence suppression can be negotiated via SDP. Refer to the User's manual for more details.

54. A new parameter 'MEGACO_MID' was added to the *ini* file. This parameter will be used in the header of the MEGACO command, instead of the local IP and port. If this parameter is not defined in the *ini* file, the default local IP and port will be used. The syntax of this parameter should be according to the 'Mid' definition of the MEGACO. If the syntax is wrong, an error message will be sent to the SysLog, and the local IP and port will be used. This is relevant for TEXT MEGACO only. Note that this parameter is NOT supported via Web or SNMP.

55. A new profile bit was added to the existing ini file parameter 'MGCPCompatibilityProfile'. Adding the value 16 to the current value activates it. For example, using the default profile 2, with full SDP support and this profile, the total value will be 26. (2 + 8 + 16). This profile controls the following:

    - In the serviceChange request, the Timestamp parameter is omitted.

    - The audit command on ROOT termination with packages descriptor returns the total supported packages for the board.

    - The default packetization period (ptime) for the transparent coder is 10 milliseconds. Using the SDP attribute ptime of course can change this.

    - The packetization period of the FAX Bypass mode will be the same as the one required for voice. When not using this profile, the packetization period will be set according to *ini* file:
      FaxBypassBasicRTPPacketInterval.

56. The event 'nt/netfail' is now supported. It will be sent if the RTP stream was broken.

1.8 Embedded Web Server New Features

1.8.1 Web Server Features

57. User help - when delaying for more than 1 second on the parameter name, a short description of this parameter is displayed.

58. Invalid parameter value warning alert – invalid parameters are colored red and a short warning message is displayed.

59. HTTP port can be configured via *ini* file.

60. The User can now retrieve the board configuration in *ini* file format. The *ini* file is downloaded from the board via the Web browser and contains all parameters that are different from their default value. The *ini* file can then be uploaded to a second board in order to apply the same configuration.

61. Restore default button – before uploading an *ini* file to the board, the User can choose to set all configuration parameters to their default values by pressing the "Restore Defaults" button. If an *ini* file is loaded without restoring the defaults first, only the parameters that appear in the *ini* file are modified.

62. Support for Date and Time configuration.

63. Message log page – adds the option to watch the error logs directly without an external SysLog server.

64. Message log page – adds the option to watch the error logs directly without an external SysLog server.

65. Save Configuration button - burning the current configuration to the flash memory without resetting the board. This action should be done before activating traffic in the board.

66. New administration features:

- Generic *ini*-Parameters page for configuration of special parameters that are NOT supported by the regular Web pages. Users can enter the parameter name or select the parameter from a list.
- Logo images upload - the User has the option to change the logo that appears in the Web pages.
- Error control page - the User can define criteria for filtering error messages that are sent to different interfaces (e.g., by severity).
- Web Command-Shell support for emulation of analog calls (on/off-hook, DTMF dialing) and recording of the signals played by the PBX. This enables the User (or administrator) to record and analyze the call-progress tones played by the PBX in order to define a suitable CPT dat file.

1.8.2 Web Server Interoperability

The HTTP connection to AudioCodes' boards was tested using the following browsers:

67. Microsoft™ Internet Explorer™ – Ver 5.0 or newer

68. Netscape™ Navigator™ – Ver 7.0 or newer

1.9 SNMP New Features

69. New parameters were added to AudioCodes proprietary acBoardMIB (more details can be found in the MIB):
    - Additional MEGACO parameters were added.
    - Added 'Set default' functionality.
    - Supplementary Parameters can be added (for set actions only). The *ini* file syntax is used: "<Parameter>=<Value>".
    - Channel Defaults – new parameters added.
70. The trap 'LineStatusChangeTrap' is now enabled by default, causing 'LineStatusChangeTrapEnableDefault' to be obsolete.
71. The following configuration files names are supported (as read-only) – FXS/FXO Coefficients, Call-Progress Tones, Voice Prompts
72. The field DSP Template Version is changed from Read-only to Read-Write.
73. New field: Enable SysLog.
74. New Table: SNMPManagers – allowing online configuration of up to three Managers that will receive SNMP Traps.
75. Community Strings for Get & Set are configurable via Ini file parameter SetCommunityString. The same string (up to 20 characters) is used for Set and for Get.
76. New SNMP Traps:
    - General Fatal Error
    - Configuration Error
77. Additional information is sent with each Trap: Severity, Probable cause, Textual description, and more.

1.10 Miscellaneous New Features

78. Customer Key – By using the new acSetCustomerKey(), acGetCustomerKey() and acEV_CUSTOMER_KEY_RESPONSE APIs, the User can burn a certain key on the board's flash and read it later. This can be useful, for example, for customers who want to tag their boards with an encrypted key for commercial purposes.
79. Reset Reason information at board start up. This debug event is issued upon completion of board start-up. It informs the User of the type of reset that caused the board to restart.
80. Time of day. This feature allows the board to keep the current date and time (day/month/year, hour/min/sec). The User can set the date and time either using acRemoteOpenBoard - for VoPLib Users (in this case the time is taken from the PC), or via the Web interface. The time is included in board errors and exceptions, and Users can also view it via the Web.

1.11 Documentation Changes

81. The old VoPLib Reference Library User's Manual, Document #: LTRT-00744 is condensed and replaced in Release 4.2 with the following two, more focused, manuals:

82. VoPLib API Reference Manual, Document #: LTRT-00840 which includes:

- API Functions (including description)
- Structures (mostly without description)
- Enumerators (mostly without description)
- Events (mostly without description)

This is an HTML document that is also available as a pdf.

83. VoPLib User's Manual", Document # LTRT-00844 which includes:

- Product description
- Control protocols overview
- Configuration Methods
- Software Installation (Host)
- Board/Channel Configuration parameters
- Using the VoPLib (API)
- Feature descriptions (divided into subjects VoPLib, PSTN, IPmedia, ATM, etc.) – main section
- VoPLib Events description
- VoPLib Error Codes
- Introduction to SS7
- V5.2 Protocol
- TPNCP
- BootP/DHCP Configuration
- Appendices
- PCI Driver related Software Installation Section was transferred from the Product User's Manuals

84. MGCP and MEGACO related information was transferred from the old VoPLib Reference Library User's Manual to the individual Product User's Manuals.

2 Known Constraints

Note: After loading and burning a CMP file to the boards FLASH memory, Users must reload all other downloadable data files (Call Progress Tones file, Voice Prompts file, Coefficients etc.). This is due to the fact that in the CMP loading and burning process all the data files that were previously stored on FLASH are erased.

2.1 New-Hardware Constraints

None reported.

2.2 Analog Interface Constraints

1. MP-124 rev A and MP-124 rev B do NOT support the following:
   - Long haul.
   - Caller ID generation.
   - MWI generation.
2. On FXO, polarity reversal detection while using 12 kHz coefficients file ("MP1xx12-1-12khz-fxo") is not functional.

2.3 Host API Constraints

3. On MP-102, for TPNCP Users, the two analog ports are mapped to TPNCP CIDs 2 and 3.
4. For additional host API constraints, which are not MP-specific, refer to AudioCodes.

2.4 IP Networking Constraints

5. In some cases having spanning tree enabled on the external Ethernet switch port connected to the board, causes the external switch to block any traffic coming in and out of the board for some time after the board is reset.
   This may cause the lost of important packets such as BootP and TFTP requests which in turn may cause the board to fail to start up.
   A possible workaround is to set the BootPRetries ini file parameter to 5, causing the board to issue 20 BootP requests for 60 seconds.
   A second workaround would be to disable the spanning tree on the port of the external switch which is connected to the board.
6. Configuring the board to auto-negotiate mode while the opposite port is set manually to full-duplex (either 10 Base-T or 100 Base-T) is invalid. It is also invalid to set the board to one of the manual modes while the opposite port is configured differently.
   The User is encouraged to always prefer full-duplex connections over half-duplex, and 100 Base-T over 10 Base-T (due to the larger bandwidth).

2.5 VoIP Engine Constraints

7. NetCoder is no longer supported on the MP-1xx product line.

8. Modem relay is not supported.

9. FRF.11 Fax and DTMF proprietary relay is no longer supported. Due to that, the channel parameter UseT38orFRF11 and the enumerator values of MF / DTMFTransportType - acProprietaryDTMFRelay / acProprietaryMFRelay are obsolete. The default values of the DTMF & MF Transport Types were changed from proprietary FRF.11 to RFC 2833.

10. (RTCP port) = (RTP port) + 1, must be kept for the incoming traffic, otherwise the RTCP traffic is ignored.

11. RFC 2198 redundancy mode with RFC 2833 is not supported (that is, if a complete DTMF digit was lost, it is not reconstructed). The current RFC 2833 implementation does support redundancy for inter-digit information lost.

12. With the channel parameters, when using FaxTransportType = TransparentWithEvents, the Fax events parameters regarding the side of the fax call (answering or calling) and the number of pages are invalid.

13. When Continuity Test Tones (COT) and/or Special Intercept Tones (SIT) detectors are enabled, the number of Call Progress Tones that can be configured is reduced. For more information refer to Section 2.4.20 “Continuity Test Tones (COT) Support” and Section 2.4.19 “Special Information Tones” in the AudioCodes VoPLib User’s Manual, Document #: LTRT-00844.

14. A channel should NOT be reopened during IBS detection. Such an action could cause the detector to send multiple events on same IBS input.

15. The resolution of the duration of digits On and Off time when dialing to the network using RFC 2833 relay, is depending on the basic frame size of the coder being used.

16. Incoming CNG T.38 packets does not switch the channel to T.38 mode

17. When CNG detector is NOT Transparent, a CNG tone received from the TDM can’t be detected using the Call Progress Tone detector.

18. Voice prompts storage space in the non-volatile memory is limited and depends on the size of the stored software (for details please contact AudioCodes).

19. The number of RTP payload packed in a single packet (VoiceSettings.M channel parameter) is limited to 5. For G.729 (20 msec) the limitation is 2.

20. When a channel is opened with many multiple RTP listeners (>15), VoiceSettings.M is limited to 2.

21. RTP payload packing of 1 is used during in-band DTMF signals, regardless of the VoiceSettings.M setting.

22. When using the TransparentWithEvents Fax & Modem Transport Types, the state machine has been changed so that on AnsAM signal an acEV_DETECT_ANSWER_TONE event is issued instead of acEV_DETECT_MODEM. On modems that start with AnsAM signal, acEV_DETECT_MODEM event will be issued only after 12 seconds of pure modem call.

2.5.1 Caller ID limitations:

23. The maximal allowed size of the Caller ID message that can be detected is according to the following formula:

[152 – (Number_Of_Parameters * 4)]/2.

24. The following fields are mandatory when generating ETSI Caller ID:

- Date.
- Time.
- All the fields of the ETSI Network Operation Use section.

2.6 MGCP Constraints

25. **MGCP** and **MEGACO** control protocols cannot coexist on the device at the same time.

26. Only one MGCP action can be associated with a requested event, (e.g., R: hd(N,A) is NOT supported).

27. DNS support - MGCP notified entity does NOT support conversion of URL.

28. On MP-102 the following ini file parameters are NOT supported:

- MGCPEndpointNumberingOffset.
- MGCPActiveEndpoints.

29. On MP-102, the use of AUEP command with wildcard (“*”) is not supported.

30. On MP-102, the first RTP channel uses UDP port of BaseUDPPort+20.

2.7 MEGACO Constraints

31. ModemDescriptor and MuxDescriptor are NOT supported in this version.

32. The auditCapabilities reply does NOT return the parameter values.

33. Version legality is NOT checked.

34. Call Agent legality is NOT checked – every request is performed.

35. In the FXS modules, only Bellcore CallerID is supported for signal generation.

36. On MP-102, the first RTP channel uses UDP port of BaseUDPPort+20.

37. Input gain and echo canceller parameters cannot be changed on the fly using MEGACO. The target termination should first be subtracted for the changes to take effect.

38. The DigitMap field in the Web/SNMP can not be changed on-line. A reset is needed for a new value.

39. If the values of input gain and echo canceller are changed via Web/SNMP, it will not affect the next call, only the one after it.

40. The IP address of call agent can not be set through BootP application anymore.

41. Size Limitations:

- Maximal command length = 6000 bytes

- Maximal number of transactions per message = 10
- Maximal number of actions per transaction = 150
- Maximal number of command requests per action = 10
- Maximal number of command replies per action = 100
- Maximal number of signals in a signal descriptor = 1
- Maximal number of signals in a signal list descriptor = 30
- Maximal number of reported signals = 50
- Maximal number of events in an event descriptor = 16
- Maximal number of reported events = 50
- Maximal number of observed events = 10
- Maximal number of event parameters = 5
- Maximal number of audit return parameters = 100
- Maximal number of DigitMap alternatives = 52
- Maximal length of DigitMap = 1256
- Maximal dial string length = 30
- Maximal length of DigitMap name = 30

2.8 Web Server Constraints

42. When using MGCP, the color indication of an open or closed channel in the channel status page in the Web is wrong (the textual information is correct).

43. The option to download a new software image via TFTP was removed from the Web server.

2.9 SNMP Constraints

44. Only one SNMP manager can access the boards/modules at one time.

45. On MP-102 the RTP MIB is NOT functional.

46. Changing some MGCP Default parameters (e.g., Voice volume) via SNMP is NOT supported, as it is possible to change them dynamically through MGCP commands on a call-by-call basis.

2.10 Miscellaneous Constraints

47. The *ini* file parameter “DePopulatedChannelsNumber ” is obsolete and is ignored.

3 Recent Revision History

3.1 Revision 4.0

3.1.1 Hardware Supported

- This version supports the following hardware platforms:
  - MediaPack MP-124/FXS, 24 analog FXS interfaces
  - MediaPack MP-108/FXS, 8 analog FXS interfaces
  - MediaPack MP-108/FXO, 8 analog FXO interfaces.
  - MediaPack MP-104/FXS, 4 analog FXS interfaces
  - MediaPack MP-104/FXO. 4 analog FXO interfaces.
  - MediaPack MP-102/FXS, 2 analog FXS interfaces

- Note that the following hardware platforms are no longer supported:
  - MediaPack MP-100
  - MediaPack MP-200

3.1.2 Analog New Features

- Life line support only on the MP-108/FXS and MP-104/FXS using unused pins on port #4, and on the MP-102/FXS using port #2.
  Life line is activated when network and power supply are lost according to "LIFELINETYPE" parameter. The "LIFELINETYPE" *ini* file parameter chooses the mode of operation of the lifeline. When set to zero (default) only power-down condition activates the Life Line. When set to one, both power down and/or LAN failure activate the life line.
- Watchdog mechanism was added. The system performs a self reset upon software failures.
- Default distinctive ring (ring #0) when no external file is loaded, 20 Hz ringing frequency, and 200/400 cadence.
- CallerID to Distinctive ringing. The User can now generate a CallerID message during the generation of distinctive ringing.
- Support for climate control on MP-124. This feature saves power by turning the fan off when the temperature is lower than a given threshold.
- The *ini* file parameters FXSCOEFFILENAME and FXOCOEFFILENAME names were changed to FXSLOOPCHARACTERISTICSFILENAME and FXOLOOPCHARACTERISTICSFILENAME respectively. The old names are still

supported for backward compatibility.

3.1.3 Host Related New Features

- Support for Sun™ Solaris™ Sparc™ 64 bit systems in TPNCP (using Forte compiler).
- Support for Windows™ XP.
- New event filtering mechanism that prevents the board from sending selected events. The User can select the desired/undesired events by *ini* file (new parameters: DISABLETPNCPEVENT) or by new API functions: acEnableTPNCPEvents() and acDisableTPNCPEvents(). By default, all events are sent from the board.
- New file "voplibapi.h". This file includes all the application necessary header files. This way the User's application does not need to include, as before, many VoP header files, just this one.
- VoP Lib sources were separated into two directories. The interface files are located under "include" directory and the rest of the files are located under "src" directory.
- New mechanism that verifies that the VoPLib version is compatible with boards firmware version. If the two are not compatible the following debug event message is sent from the board/VoP Lib: "TPNCP version error…".
- The following new analog events were added: acEV_ANALOG_IF_FXO_STATUS, acEV_ANALOG_IF_FXS_STATUS. These two events are replacing the previous unified event acEV_ANALOG_IF_STATUS.
- Addition of CallerID to the distinctive ringing mechanism. New parameters were added to acAnalogIFGenerateDistinctiveRing() API function: EnableCallerIDMessage and CallerIDMessage.
- New acOpenRemoteBoard() overloading. With this new function the User can decide whether to supply the board parameters or not. If board parameters are supplied (non NULL value), then both the VoPLib and the board use these parameters. If the board parameters are not supplied, then the VoPLib is configured automatically from the board.
- New acGetInfrastructuresParameter() API function for obtaining various data from the board. Currently the data that can be requested are the board parameters.
- New parameter "UserDialTiming" was added to acPlayCallProgressTone(), acDial(),and acPlayVoicePrompt() API functions. This parameter determines the order of playing of the CPT, digits and voice prompts, when these functions are called one after the other. Refer to the VoP manual for more details.
- Add VoicePrompt on-the-fly. It is now possible to use acAddVoicePrompt when channels are opened on the board. Formerly acAddVoicePrompt had to be issued prior to opening any channel on the board.
- The following new RTP related events were added:
  - acEV_BROKEN_RTP_CONNECTION – this event is issued when for more than 300 msec there was no reception of RTP packets on a specific channel.
  - acEV_RTP_CONNECTION_REASTABLISHED – this event is reported when the RTP reception resumes.

- acEV_RTCP_CLOSE – this event is issued when closing channel. This event includes statistics for the closed channel (only relevant to RTP channels).

- Delete all Voice Prompts. It is now possible to delete all Voice Prompts using command acDeleteAllVoicePrompts().A new (optional) parameter int *EventInfoSize was added to the function acGetEvents(). This parameter (if used) is a pointer in which the actual EventInfo size is returned.

- A Tool Command Language (TCL) extension implementing the interface to the VoPLib was added to the example folder.

- The numbers of the following error codes were changed:

  - CMD_NO_LONGER_SUPPORTED = 0x36100 was changed to:
    CMD_NO_LONGER_SUPPORTED = 0x370

- From this version on, most internally used functions (not API functions) get the prefix "_".

3.1.4 Ethernet Interface-Related New Features

- New API function – acChangeIPAddress(). This API enables the User to change the boards IP address while the board is running.

- New heartbeat mechanism. The User can configure the board to send a special heartbeat packet in User-defined intervals. The destination IP address and port for the heartbeat packets are also definable. Three new *ini* file parameters are used to configure the heartbeat mechanism:
  HeartBeatIntervalmSec HeartBeatDestIP and HeartBeatDestPort.

- Added support to IP Multicasting level1 according to RFC 1112.

3.1.5 VoIP Engine New Features

- Support for Fax Bypass RTP Dynamic payload type. This enables the Fax Bypass feature to work with other similar modes on other vendors' Gateways.

- A new feature was added that sends only RFC 2833 RTP packets, while receiving any types of packet.

- Support for the IETF RTP Payload for Comfort Noise (CN) per RFC 3389.

- Enhancement of the Voice Prompt playback mechanism. Voice Prompts that are recorded with G.711 A-law and G.711 µ-law can be played to a channel being opened with any coder (e.g., G.723). Voice Prompts that are being recorded with anything other than G.711 A-law and G.711 µ-law can be played only to the TDM side when the channel is being opened with the same coder as the Voice Prompts coder

- New event EV_RTCP_CLOSE is issued when closing channel. This event includes statistics for the closed channel (only relevant to RTP channels).

- Support for detection of call progress tones and DTMF from the packet network side. The User determines on which side does the channel detect CPT and DTMF using the parameter IBSDRedirection in the channel parameters.

- Support G.729 10 msec packet interval mode.

- RTP Dynamic Payload Types according to RFC 2833 (DTMF Relay) and RFC 2198 (RTP Redundancy).
- Immediate opening after closing a channel capability. In previous versions, Users had to wait 100 msec between the time a channel was closed and reopened; now this delay is not required.
- New DTMF transport type: "RFC 8233 with decoder muting". This mode allows for generation of DTMF relay packets towards the IP network and blocking of DTMF relay packets received from the IP network.
- Support for changing Input Gain of a channel on the fly (using acChangeChannelParameters).
- Support for Type 1 and Type 2 Caller ID detection (on FXO devices) and generation (on FXS devices) according to Bellcore standard (previously only Bellcore type 1 was supported). On FXS devices, support for the generation of visual Message Waiting Indication (MWI) for both FSK transmission and voltage.
- On FXO devices, support for visual MWI detection.
- New acStopCallerID() API function. Stops Caller ID generation.
- New mechanism for a well defined order-of-play of call progress tones, digits and voice prompts. A new UserDialTiming parameter was added to acPlayVoicePrompt(), acDial() and acPlayCallProgressTone(). This parameter allows the User to choose between immediate, concatenated or default playing of the new Voice Prompt/Digit/CPT.
- Improvement of RTCP jitter calculation. The jitter statistics are now accurate to 1 msec.
- Support for adding voice prompts on-the-fly. Refer to Section 1.3 Host Related New Features on page 8.

  The DSPVersionTemplateNumber parameter selects the DSP firmware and hence the supported coder group. The following list specifies the different values for DSPVersionTemplateNumber and their respective supported coder list:

  - 0 - DSP firmware supports PCM/ADPCM, G723, G729A, GSM.
  - 1 - DSP firmware supports PCM/ADPCM, NetCoder.
  - 2* - DSP firmware supports PCM/ADPCM, G723, G729A, GSM.
  - 3* - DSP firmware supports PCM/ADPCM, NetCoder

(* These templates include an additional IPmedia features: Answer Detector.)

3.1.6 SNMP Support

- RTP MIB - The RTP MIB is supported per RFC 2959. It contains objects relevant to the RTP streams generated and terminated by the board and to the RTCP information related to these streams
- acBoardMIB - This AudioCodes proprietary MIB contains objects related both to the configuration of the board and channels as well as to run-time information.
- New SNMP board related parameters were added

- DisableSNMP *ini* file parameter – This parameter disables the SNMP.
- SNMPPort *ini* file parameter – this parameter specifies the port number for requests (from the manger to the agent) and responses (from agent to manger).
- SNMPTrapPort *ini* file parameter - specifies the port number on which the traps are transmitted (from agent to manager).

3.1.7 MGCP New Features

- Users should provision the *ini* file parameter "MGCControlProtocolType" for using the various standard control protocols:
  0 – None
  1 – MGCP
  2 – MEGACO
  3 – Reserved.
- New 'Global Persistent Events' engine. Allows Users to define additional persistent events other than the ones defined in RFC 2705. Persistent events can be provisioned via *ini* file using MGCPPERSISTENTEVENTS parameter.
- New AudioCodes Proprietary extension x-MAC: Used for sending the board's MAC address during RSIP commands. This new feature is enabled when *ini* file parameter 'MGCPSendMACWithRSIP' is set to '1'.
- An MGCP RSIP command can be used to remotely reset the device (Prop. Implementation of RSIP).
- Transaction ID start value and range provisioning via *ini* file parameters TransactionIDBase and TransactionIDRange.

3.1.8 MEGACO Support

Following are the packages supported (Users should refer to the User's Manual for details regarding each product).

- The TrunkPack series MEGACO client supports the basic set of packages, as defined in Annex E of the RFC 3015:
  - Generic Media Package: g
  - Base Root Package: root
  - Tone Detection Package: tonedet
  - Tone Generator Package: tonegen
  - DTMF Generator Package: dg
  - DTMF Detection Package: dd
  - Call Progress Tones Generator Package: cg
  - Call Progress Tones Detection Package: cd

- Analog Line Supervision Package: al
- Basic Continuity Package: ct
- Network Package: nt
- RTP Package: rtp
- TDM Circuit Package: tdmc
- Basic Announcements Package: an
- Expanded Call Progress Tones Generator Package: xcg
- Basic Service Tones package: srvtn
- Expanded Service Tones generation Package: xsrvtn
- Enhanced Alerting Package: alert
- Analog Display Signaling Package: andisp - (caller ID)
- Extended Analog Line Supervision Package: xal
- FXO Package fxo
  Note that the remote hangup event in this package is NOT supported

- DNS support – The Call agent can be defined using a domain name instead of IP address.
- ServiceChange can be sent from the MGC, to put endpoint/s in or out of service.
- In addition to the text format (ABNF) encoding, there is support for binary encoding (ASN.1)
- Support for dynamic payload assignment in the SDP. Support for the ComfortNoise payload type (13).
- MGCP and MEGACO cannot coexist for controlling the same TrunkPack

3.1.9 Web Browser Support

3.1.9.1 Web Browser Features

- The Web server was upgraded in the board/module software, providing the ability to set up board parameters using a standard Web interface. The following new features were added:
  - "Quick board Setup" menu. This menu Guides the User through mandatory steps and basic configuration of the board.
  - Retrieve from local disk and DOWNLOAD *ini* file and configuration (CPT, COEFF, CAS, VPMT) via HTTP.
  - DOWNLOAD of run time software via HTTP (RAM.CMP)

- Improved help menus. Help related to each page with parameters explanations and usage.
- Enhanced information about software revisions, hardware configurations, MAC address etc.

- Embedded Web server password protection support was added. The User can change the password from the “Change Password” Web page (note that a hardware restart must follow this operation). The User can “reset” the password to the default value of “Admin” by using the *ini* file parameter “RESETWEBPASSWORD” (this should always be applied the first time the hardware is used). If this parameter is not part of the *ini* file, the hardware remembers the last password used. The default Username is ***Admin*** and the Password is ***Admin***.
- The analog coefficients file loading using the Web interface, is not supported in the MP-10X/FXO boards.

3.1.9.2 Web Browser Interop

The Web browser connection to AudioCodes' boards was tested using the following browsers:

- Microsoft™ Internet Explorer™ – Ver 5.0 or newer
- Netscape™ Navigator™ – Ver 7.0 or newer

3.2 Previous Releases

Details of previous releases can be found in the Release Notes of Version 4.0 Document # LTRT-00616, published by AudioCodes on Oct-21-2002.

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